IN THE CLAIMS

Claim 1 (previously presented): A graphametric equalizer comprising:

a plurality of equalizing filters spanning a predetermined audio bandwidth;

a data processor;

a data input device in communication with the data processor;

a translation function algorithmic software directing the data processor;

a softening function algorithmic software directing the data processor; and

a data storage unit, wherein discrete center frequency data, discrete bandwidth data and discrete gain data is stored and supplied to the data processor such that the data processor, directed by the translation function algorithmic software, can determine filter parameters using algorithmically defined relationships among the discrete center frequency data, discrete bandwidth data and discrete gain data such that the plurality of equalizing filters can be recharacterized by the filter parameters, and further wherein discrete timing data and discrete incrementing data is stored and supplied to the data processor such that the data processor, directed by the softening function algorithmic software, can determine gain incrementing parameters and timing parameters such that the plurality of equalizing filters can be recharacterized.

Claim 2 (original): The graphametric equalizer according to claim 1 wherein each equalizing filter comprises an allpass filter having a multiplier dependent upon a predetermined bandwidth and a predetermined peak gain for a cut region below 0 dB and further dependent solely upon a predetermined bandwidth for a boost region above 0 dB.

Claim 3 (previously presented): A graphametric equalizer comprising:

a plurality of equalizing filters spanning a predetermined audio bandwidth;

a data processor;

a data input device in communication with the data processor;

a translation function algorithmic software directing the data processor;

a softening function algorithmic software directing the data processor; and

a data storage unit, wherein discrete center frequency data, discrete bandwidth data and discrete gain data is stored and supplied to the data processor such that the data processor, directed by the translation function algorithmic software, can determine filter parameters using algorithmically defined relationships among the discrete center frequency data, discrete bandwidth data and discrete gain data such that the plurality of equalizing filters can be recharacterized by the filter parameters, and further wherein discrete timing data and discrete incrementing data is stored and supplied to the data processor such that the data processor, directed by the softening function algorithmic software, can determine gain incrementing parameters and timing parameters such that the plurality of equalizing filters can be recharacterized;

wherein each equalizing filter comprises an allpass filter having a multiplier dependent upon a predetermined bandwidth and a predetermined peak gain for a cut region below 0 dB and further dependent solely upon a predetermined bandwidth for a boost region above 0 dB; and

wherein the translation function algorithmic software is configured to approximate the allpass filter multiplier in the cut region below 0 dB and comprises the multiplier approximation function:

$$\beta \approx \tan(\pi BW/F_s)/\log_2(3)\cdot k \text{ ; wherein}$$

β is the allpass filter multiplier, BW is a user selected allpass filter bandwidth, Fs is a user selected sampling frequency, and k is a user selected peak gain of the equalizing filter.

Claim 4 (previously presented): A graphametric equalizer comprising:

a plurality of equalizing filters spanning a predetermined audio bandwidth;

a data processor;

a data input device in communication with the data processor;

a translation function algorithmic software directing the data processor;

a softening function algorithmic software directing the data processor; and

a data storage unit, wherein discrete center frequency data, discrete bandwidth data and discrete gain data is stored and supplied to the data processor such that the data processor, directed by the translation function algorithmic software, can determine filter parameters using algorithmically defined relationships among the discrete center frequency data, discrete bandwidth data and discrete gain data such that the plurality of equalizing filters can be recharacterized by the filter parameters, and further wherein discrete timing data and discrete incrementing data is stored and supplied to the data processor such that the data processor, directed by the softening function algorithmic software, can determine gain incrementing parameters and timing parameters such that the plurality of equalizing filters can be recharacterized;

wherein the translation function algorithmic software is further configured to form a reciprocal estimate for a constant x and comprises the reciprocal estimate function:

$$1/x \approx (1/s) * 2^{-n-2} - r * 2^{-2n-1} + 2^{-n-1} \text{ ; wherein}$$

s is a scaling parameter that will ordinarily have a value between about 0.5 and about 0.6 and is a constant ~~for each application~~, and wherein the value of n is represented by the most significant digit of the constant x, and further wherein $r = x - 2^n$.

Claim 5 (currently amended): A digital graphametric equalizer comprising:

a plurality of equalizing filters spanning a predetermined audio bandwidth, each equalizing filter comprising an allpass filter having a variable multiplier dependent upon a desired bandwidth and a desired peak gain for a cut region below 0 dB and further dependent solely upon a desired bandwidth for a boost region above 0 dB; and

a translation algorithmic software configured to generate a reciprocal estimate for the desired peak gain k and that comprises the reciprocal estimate function:

$$1/k \approx (1/s) * 2^{-n-2} - r * 2^{-2n-1} + 2^{-n-1} \text{ ; wherein}$$

s is a scaling parameter that will ordinarily have a value between about 0.5 and about 0.6 and is a constant ~~for each application~~, and wherein the value of n is represented by the most significant digit of the desired peak gain k, and further wherein $r = k - 2^n$.

Claim 6 (currently amended): The graphametric equalizer according to claim 5 wherein the translation algorithmic software is further configured to approximate the allpass filter variable multiplier in the cut region below 0 dB and comprises the multiplier approximation function:

$$\beta \approx \tan(\pi BW/F_s)/\log_2(3)\cdot k \text{ for } \tan(\pi BW/F_s) \leq \cancel{\text{about}}\ 0.0625 \text{ ; wherein}$$

β is the allpass filter multiplier, BW is a predetermined allpass filter bandwidth, Fs is a sampling frequency, and k is a predetermined peak gain of the equalizing filter.

Claim 7 (currently amended): The graphametric equalizer according to claim 5 wherein the translation algorithmic software is further configured to approximate the allpass filter variable multiplier in the cut region below 0 dB and comprises the multiplier approximation function:

$\beta = [\tan(\Omega/2)-1]/[\tan(\Omega/2)+1]$ for $\tan(\pi BW/F_s) >$ ~~about~~ 0.0625 ;
wherein

β is the allpass filter multiplier, BW is a predetermined allpass filter bandwidth, Fs is a predetermined sampling frequency, and k is a predetermined peak gain of the equalizing filter.

Claim 8 (previously presented): A graphametric equalizer comprising:

a plurality of digital equalizing filters spanning a predetermined audio bandwidth, each equalizing filter comprising an allpass filter having a variable multiplier parameter that is dependent upon a desired bandwidth and a desired peak gain for a cut region below 0 dB and further that is dependent solely upon a desired bandwidth for a boost region above 0 dB;

translating means for translating a desired bandwidth and a desired peak gain and generating the variable multiplier parameter such that the plurality of digital equalizing filters can be recharacterized with a desired multiplier; and

softening means for timing user inputs and incrementing filter parameters such that the plurality of digital equalizing filters can be recharacterized.

Claim 9 (currently amended): A graphametric equalizer comprising:

a plurality of digital equalizing filters spanning a predetermined audio bandwidth, each equalizing filter comprising an allpass filter having a variable multiplier parameter that is dependent upon a desired bandwidth and a desired peak gain for a cut region below 0 dB and further that is dependent solely upon a desired bandwidth for a boost region above 0 dB;

translating means for translating a desired bandwidth and a desired peak gain and generating the variable multiplier parameter such that the plurality of digital equalizing filters can be recharacterized with a desired multiplier; and

softening means for timing user inputs and incrementing filter parameters such that the plurality of digital equalizing filters can be recharacterized,

wherein the translating means comprises an algorithmic software configured to generate a reciprocal estimate for the desired peak gain k using the relationship:

$$1/k \approx (1/s) * 2^{-n-2} - r * 2^{-2n-1} + 2^{-n-1} \text{ ; wherein}$$

s is a scaling parameter that will ordinarily have a value between about 0.5 and about 0.6 and is a constant ~~for each application~~, and wherein the value of n is represented by the most significant digit of the desired peak gain k, and further wherein $r = k - 2^n$.

Claim 10 (currently amended): The graphametric equalizer according to claim 9 wherein the translating means further comprises an algorithmic software configured to generate the allpass filter variable multiplier parameter in the cut region below 0 dB and comprises the multiplier approximation function:

$$\beta \approx \tan(\pi BW/F_s)/\log_2(3)\cdot k \text{ for } \tan(\pi BW/F_s) \leq \cancel{\text{about}}\ 0.0625\text{; wherein}$$

β is the allpass filter multiplier, BW is a predetermined allpass filter bandwidth, Fs is a predetermined sampling frequency, and k is the peak gain of the equalizing filter.

Claim 11 (previously presented): A graphametric equalizer comprising:

a plurality of digital equalizing filters spanning a predetermined audio bandwidth, each equalizing filter comprising an allpass filter having a variable multiplier parameter that is dependent upon a desired bandwidth and a desired peak gain for a cut region below 0 dB and further that is dependent solely upon a desired bandwidth for a boost region above 0 dB;

translating means for translating a desired bandwidth and a desired peak gain and generating the variable multiplier parameter such that the plurality of digital equalizing filters can be recharacterized with a desired multiplier; and

softening means for timing user inputs and incrementing filter parameters such that the plurality of digital equalizing filters can be recharacterized,

wherein the softening means is configured to limit equalizer gain variations to a range of about 0.05 to about 0.06 in linear space no more than once for every 64 samples for a sampling rate of about 44.1 kHz.

Claim 12 (original): The graphametric equalizer according to claim 11 wherein the softening means is further configured to limit equalizer parameter variations to no more than a single modification for every 64 input samples for a sampling rate of about 44.1 kHz.

Claim 13 (currently amended): A method of digital equalizer control comprising the steps of:

providing an allpass filter-based equalization filter having a variable parameter multiplier;

receiving a user-selected gain and user-selected bandwidth for an allpass filter associated with the equalization filter;

generating a reciprocal estimate for the user-selected gain using the relationship:

$$1/k \approx (1/s) * 2^{-n-2} - r * 2^{-2n-1} + 2^{-n-1}$$; wherein

s is a scaling parameter that will ordinarily have a value between ~~about~~ 0.5 and ~~about~~ 0.6 and is a constant ~~for each application~~, and wherein the value of n is represented by the most significant digit of the desired peak gain k, and further wherein $r = k - 2^n$;

translating the user-selected gain and user-selected bandwidth into a desired first multiplier parameter β for the allpass filter via a multiplier approximation function expressed as:

$$\beta \approx \tan(\pi BW/F_s)/\log_2(3)\cdot k \text{ for } \tan(\pi BW/F_s) \leq \text{about } 0.0625\text{; wherein}$$

β is the allpass filter multiplier, BW is a predetermined allpass filter bandwidth, Fs is a predetermined sampling frequency, and k is the peak gain of the equalizing filter; and

processing a desired input signal via the allpass filter such that the equalizing filter can achieve a negative gain in a predetermined cut region below 0 dB.

Claim 14 (currently amended): The method according to claim 13 further comprising the step of translating the user-selected gain and user-selected

bandwidth into a desired second allpass filter multiplier parameter β for the allpass filter via a multiplier approximation function expressed as:

$$\beta = [\tan(\Omega/2)-k]/[\tan(\Omega/2)+k];\ \text{wherein}$$

Ω is the user-selected bandwidth, such that the equalizing filter can achieve a negative gain for a desired input signal in a predetermined cut region below 0 dB when $\tan(\pi BW/F_s) >$ ~~about~~ 0.0625.

Claim 15 (original): The method according to claim 13 further comprising the step of receiving a user-selected center frequency for the equalizing filter associated with the allpass filter and translating the user-selected center frequency into a second variable allpass filter multiplier parameter such that the equalizing filter can be realized via a second-order allpass filter.

Claim 16 (currently amended): The method according to claim 13 further comprising the step of translating the user-selected bandwidth into a desired allpass filter second multiplier parameter β for the allpass filter via a multiplier approximation function expressed as:

$$\beta = [\tan(\Omega/2)-1]/[\tan(\Omega/2)+1];\ \text{wherein}$$

Ω is the user-selected bandwidth, such that the equalizing filter can achieve a positive gain for a desired input signal when the input signal is in a predetermined boost region above 0 dB.

Claim 17 (original): A method of digital equalizer control comprising the steps of:

providing an allpass filter-based equalization filter;

receiving a user-selected center frequency, user-selected bandwidth and user-selected gain setting;

generating a reciprocal estimate for the user-selected gain setting;

generating a first allpass filter multiplier parameter in the region where $\tan(\Omega/2) \leq$ about 0.0625 based in part upon the reciprocal estimate for the user-selected gain setting, where Ω is the user-selected bandwidth;

characterizing an allpass filter via the first filter multiplier parameter; and

processing a predetermined input signal via the equalization filter to achieve a negative gain in a desired cut region below 0 dB.

Claim 18 (original): The method of claim 17 further comprising the steps of:

timing and incrementing the recharacterization of the allpass filter via the first multiplier parameter such that undesirable audible artifacts are substantially reduced.

Claim 19 (original): The method of claim 17 further comprising the steps of:

generating a second allpass filter multiplier parameter in the region where $\tan(\Omega/2) >$ about 0.0625;

further characterizing an allpass filter via the second filter multiplier parameter; and

processing a predetermined input signal via the equalization filter to achieve a negative gain in a desired cut region below 0 dB.

Claim 20 (original): The method of claim 19 further comprising the steps of:

timing and incrementing the recharacterization of the allpass filter via the second multiplier parameter such that undesirable audible artifacts are substantially reduced.

Claim 21 (original): The method of claim 17 further comprising the steps of:

generating a third allpass filter multiplier parameter based upon the user-selected center frequency;

further characterizing the allpass filter via the third multiplier parameter; and

processing a predetermined input signal via the equalization filter to achieve a positive gain in a desired boost region above 0 dB.

Claim 22 (original): The method of claim 21 further comprising the steps of:

timing and incrementing the recharacterization of the allpass filter via the multiplier parameters such that undesirable audible artifacts are substantially reduced.